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Friedbert BERENS et al. :

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For: METHOD OF DECODING A TURBO-CODE ENCODED SIGNAL IN A RECEIVER AND CORRESPONDING RECEIVER :

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Sir:

Under the provisions of 35 U.S.C. §119, there is filed herewith a certified copy of European Application No. 02-023841.6, filed October 24, 2002, in accordance with the International Convention for the Protection of Industrial Property, 53 Stat. 1748, under which Applicants hereby claim priority.

Respectfully submitted,

Date: 10/24/03

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Der Präsident des Europäischen Patentamts;
Im Auftrag

For the President of the European Patent Office

Le Président de l'Office européen des brevets
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Bezeichnung der Erfindung/Title of the invention/Titre de l'invention:
(Falls die Bezeichnung der Erfindung nicht angegeben ist, siehe Beschreibung.
If no title is shown please refer to the description.
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Method of decoding an incident turbo-code encoded signal in a receiver, and corresponding receiver, in particular for mobile radio systems

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Method of decoding an incident turbo-code encoded signal in a receiver, and corresponding receiver, in particular for mobile radio systems

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Method of decoding an incident turbo-code encoded signal in a receiver, and corresponding receiver, in particular for mobile radio systems

The invention relates in general to channel decoding techniques, in particular to turbo-codes.

An application of the invention is directed in general to the field of wireless communication systems, and more particularly to the CDMA systems such as the different CDMA based mobile radio systems like CDMA 2000, WCDMA (Wide Band CDMA) or the IS-95 standard.

In mobile radio systems channel coding is used to make the data transmission over the mobile radio channel more robust against noise and interference. Standard channel coding schemes used in existing mobile radio systems like GSM are Convolutional Coding, Reed Salomon Coding and other kinds of block codes.

The third generation mobile radio system specifies Convolutional codes as well as Turbo-codes as channel coding techniques [3GPP, Technical Specification Group Radio Access Network ; *Multiplexing and channel coding (FDD)* ; (3G TS 25.212 version 3.5.0(2000-12)), Release 1999].

In Turbo-code encoders forward error correction is enabled by introducing parity bits. For Turbo-codes, the original information, denoted as systematic information, is transmitted together with the parity information. The encoder for 3GPP consists of two recursive systematic convolutional (RSC) encoders with constraint length K=4, which can also be interpreted as 8-state finite state machines. The first RSC encoder works on the block of information in its original, the second one in an interleaved sequence.

On the receiver side, there is a corresponding component decoder for each of them. Each component decoder implements a so-called Maximum-A-Posteriori (MAP) Algorithm, and is usually so-called Soft-in-Soft-out (SISO) decoder.

Each block is decoded in an iterative manner. The systematic information and the parity information serve as inputs of the first component decoder (MAP1). The soft-output of MAP1 reflects its confidence on the received bits of being sent either as '0' or '1'. These confidences are interleaved in the same manner as in the encoder and passed to the second component decoder (MAP2) as a-priori information. The second component decoder uses this information to bias its estimation comprising the interleaved systematic information and the parity information of the second encoder. The soft-outputs are again passed on to MAP1, and so on. The exchange continues until a stop criterion is fulfilled. Stop criteria range from simple cases, such as "fixed number of iterations", over cyclic redundancy check (CRC) to rather complex statistical analysis.

Implementation issues for Turbo-decoder architectures using the MAP algorithm have already been discussed in several papers and are well known [A.Worm. *Implementation Issues of Turbo-Decoders*. Phd thesis, Institute of Microelectronic Systems, Departement of Electrical engineering and Information Technology, University of Kaiserslautern, Forschungsberichte Mikroelektronik, Bd.3, Germany, 2001].

The MAP algorithm is transformed into the logarithmic domain to reduce operator strength [S.S. Pietrobond and A.S. Barbulescu. A Simplification of the Modified Bahl Decoding Algorithm for Systematic Convolutional Codes. In *Proc. International Symposium on Information Theory and its Applications*, pages 1073-1077, Sydney, Australia, November 1994]: multiplications become additions, and additions are replaced by a modified comparison. It consists of a forward recursion, a backward recursion and soft-output calculation.

Two MAP algorithms, well known by the man skilled in the art, work in the logarithmic domain. The first one is called LogMAP algorithm and shows the same performance than ideal MAP algorithm using exponential functions.

The second one, called MaxLogMAP algorithm, is a sub-optimum version of the LogMAP algorithm.

In principle, the performance of a LogMAP based turbo-code decoder is better than the performance of a MaxLogMAP based turbo-code decoder. However, under certain circumstances, the performance of the MaxLog MAP based decoder behaves much more robust and it exceeds the performance of the LogMAP based decoder.

In existing systems, one of the above-mentioned decoding algorithms are used, either the LogMAP or the MaxLogMAP.

The invention proposes a dynamic switching in the existing solutions so that the overall turbo-decoder according to the invention works always in the best performing mode (either LogMAP or MaxLogMAP), depending on the transmission conditions.

As claimed, the invention proposes a method of decoding an incident turbo-code encoded signal in a receiver, comprising a step of receiving said signal from a transmission channel, and a step of digitally turbo-code decoding.

According to a general feature of the invention, the method further comprises a step of dynamically determining a quality information representative of the conditions of the channel state estimation, and a step of dynamically comparing said quality information with a predetermined criteria for defining a binary result corresponding to good or bad estimation conditions.

And, the turbo-code decoding step comprises dynamically selecting a Maximum a Posteriori algorithm in the logarithmic domain, called LogMAP algorithm, in presence of good conditions, or an approximation of said Maximum a Posteriori algorithm in the logarithmic domain, called MaxLogMAP algorithm, in presence of bad conditions.

The conditions of the channel state estimation can be predicted by knowing the quality of the Signal to Interference Ratio (called SIR) estimation.

More precisely, according to an embodiment of the invention, said determination step comprises determining an error information representing an error of the SIR estimation, said error information being thus said quality information.

And, said predetermined criteria comprises at least one predetermined threshold.

Said error information can be the absolute value of the error of the SIR estimation. And, said LogMAP algorithm is selected if the absolute value of said error is smaller than said threshold, whereas said MaxLogMAP algorithm is selected if the absolute value of said error is greater than or equal to said threshold.

Practically, said threshold can be chosen so that it is smaller than or equal to about 0,5 dB, for example equal to about 0,2 dB.

One possibility for calculating said absolute value of the error of the SIR estimation consists in calculating the variance of the SIR estimation.

However, the inventors have observed it can be much simpler to predict the conditions of the channel state estimation by knowing the speed of the receiver (when the receiver is mobile) or the speed of a mobile terminal cooperating with said receiver (when the receiver is a base station for example).

In other words, according to an embodiment of the invention wherein the receiver is mobile, for example a cellular mobile phone, and belongs to a wireless communication system, said determination step comprises a mobile receiver speed estimation, said speed being said error information.

The receiver can be also a base station cooperating with a mobile terminal, for example a cellular mobile phone. In such a case, said determination step comprises a mobile terminal speed estimation and said speed being also said error information.

When the speed is considered as an error information, said predetermined criteria can comprise also the delay profile of the transmission channel. More precisely, said delay profile can be used for determining thresholds used to select either the LogMAP algorithm or the MaxLogMAP algorithm.

The invention as claimed proposes also a receiver, comprising reception means for receiving an incident turbo-code encoded signal from a transmission channel, and digital processing means connected

to the reception means and including demodulation means and turbo-code decoding means.

According to a general feature of the invention, turbo-code decoding means are controllably configurable, in response to a binary control signal, between a first configuration implementing a Maximum a Posteriori algorithm in the logarithmic domain, called LogMAP algorithm, and a second configuration implementing an approximation of said Maximum a Posteriori algorithm in the logarithmic domain, called MaxLogMAP algorithm.

And said digital processing means further comprise auxiliary processing means for dynamically determining a quality information representative of the conditions of the channel state estimation, and control means for dynamically comparing said quality information with a predetermined criteria and delivering said binary control signal having a first value corresponding to good estimation conditions and a second value corresponding to bad estimation conditions, said turbo-code decoding means being dynamically switched in their first configuration when the binary control signal has the first value, and in their second configuration when the binary control signal has the second value.

According to an embodiment of the invention, said auxiliary processing means comprise means for determining an error information representing an error of the Signal to Interference Ratio estimation, said error information being said quality information, and said predetermined criteria comprises at least one predetermined threshold.

According to an embodiment of the invention, said error information is the absolute value of the error of the SIR estimation and said control signal has the first value if the absolute value of said error is smaller than said threshold, and the second value if the absolute value of said error is greater than or equal to said threshold.

When the receiver is mobile and belong to a wireless communication system, said auxiliary processing means may comprise speed estimation means for determining the speed of the mobile receiver, said speed being said error information.

When the receiver belongs to a wireless communication system and cooperates with a mobile terminal, said auxiliary processing means may comprise speed estimation means for determining the speed of the mobile terminal, said speed being also said error information.

Other advantages and features of the invention will appear on examining the detailed description of embodiments, these being in no way limiting and of the appended drawings in which :

- figure 1 shows the structure of the two different NSC used in UMTS for Convolutional encoding,
- figure 2 shows a part of a trellis which represents the possible transitions in one time step,
- figure 1 shows an UMTS Turbo-code encoder,
- figure 2 shows a generic Turbo decoder,
- figure 3 shows a receiving chain of mobile phone
- figure 4 illustrates more in details but still diagramatically, the internal architecture of a channel decoder according to the invention including in the mobile phone according to the invention, and,
- figure 5 illustrates the way of selecting between LogMAP and MaxLogMAP algorithm depending on the speed of the mobile phone.

Before explaining in details, in particular the dynamic switching of the decoding algorithm according to the invention, we explain thereafter some general features of the two used algorithms, which are already well known by the man skilled in the art.

1 Encoding

1.1 General Considerations and Convolutional encoding

Convolutional encoding is performed by calculating the modulo-2 sum of the input values of the current and/or selected previous time steps. Implementation therefore is straightforward and mainly consists of shift register and a couple of exclusive-OR gates. Through the way those are switched, different kinds of convolutional codes can be realized:

Systematic Codes: One of the output streams is equal to the input stream, the systematic information.

Non-Systematic Codes (NSC): Each output is a parity information. Parity information is produced by taking the modulo-2 sum of shift register entries stating the history of the encoding process.

Recursive Codes: A special parity signal is produced and fed back in conjunction with the systematic input.

Non-Recursive Codes: No such feedback loop exists.

An instance of a convolutional encoder is defined by a combination of these properties, the memory depth (constraint length) and the logical functions used to produce the parity information. These properties are described through generator polynomials.

1.2 Code-Trellis

A code trellis is the unrolled state chart of a finite-state machine. The number of states the encoder can be in (N) is a function of the constraint length K:

N = 2K-1.

Depending on the nature of the code (RSC, NSC,...) only certain transitions are possible. A trellis is used to depict those transitions.

1.3 Trellis-termination

For the considered codes the initial state of the trellis is always known to be the all-zero state. Without taking any precautions, the encoder ends in an arbitrary state, leaving no hint where to start the backward recursion. This can be counteracted by driving the encoder into a defined final state. Reaching the final state (e.g. the all-zero state) can be achieved by appending a sequence, which steers the encoder towards the final state as fast as possible. This sequence is also depending on the state the encoder is in after the last information bit has been coded. The length of this sequence is equal to K-1; the transmitted bits are called tailbits.

1.4 Interleaving

Trellis based decoding is very vulnerable to burst errors. If a sequence of transmitted bits is corrupted, the decoding becomes inaccurate. Therefore a scheme to break up the neighborhood-relations is applied: the interleaving.

The key idea behind interleaving is to transmit the bits in a different order than they are produced or consumed. For instance, while bit 4 is encoded consecutively with its neighbors 3 and 5, it might be next to 312 and 1021 during transmission. A burst error in the channel effects bits 312, 4, and 1021. On the receiver side, these errors are spread again through the deinterleaver, which restores the initial order. Thus the decoding is less affected.

1.5 <u>Turbo-encoding</u>

A Turbo code encoder consists of two constituent convolutional encoders and an interleaver. The convolutional codes are fixed to be the RSC codes of rate 1/2 and generator polynomials (13,15/(octal notation) introduced before.

The systematic information of the second encoder is not transmitted because it can be reconstructed (by deinterleaving) from the systematic output of the first encoder. By this a rate of R = 1/3 is achieved. Figure 1 shows the detailed UMTS Turbo code encoder. The trellis termination leads each encoder into its final state separately. This dissolves the dependency between the systematic information of the first and second encoder for the tailbits, because these lead each encoder independent from the other by activating the respective switch, see figure 1. Hence the last six bits per encoder (systematic and parity for each) have to be transmitted separately. This results in a total overhead of 12 bits per block.

2.Decoding

Decoding convolutional codes is keeping track of the transitions that took place in the encoder. From those the input symbols which have been sent are deducted. Due to the degradations caused by the channel, only estimates of the systematic and parity bits are available, which will both be called channel values here. There are two different kinds of outputs:

Hard values: they merely indicate if a symbol is supposed to be "1" or "0".

Soft values: These also deliver a measure for the reliability of the decision (the hard decision is extend by the probability that the decision is correct).

For Turbo decoding only soft-in values are relevant. Based on the channel values, probabilities can be computed that certain combinations of systematic and parity bit occurred. From this and considering the encoder history, the probability that the encoder was in a given state at a given time-step can be computed.

Two approaches exist to deal with those state-probabilities. The maximum likelihood based Viterbi algorithm uses them to search the most likely code-word. For this it traverses the trellis from the all-zero state to the end state and looks for the most likely sequence. The states chosen for the survivor path indicate the most likely sequence of symbols that has been send. Hence a Viterbi Decoder is a sequence estimator.

The maximum-a-posteriori (MAP) algorithm on the other side estimates the probability that the encoder was in the given state and that the current state leads to the final state given the remainder of the channel values. This can be efficient computed by a forward and backward recursion over the trellis. Afterwards, for each bit the probabilities for those states associated with a systematic "0" are added and compared to those associated with a "1". The symbol with the higher probability is assumed to be the sent one. As this works on bit rather than on sequence level, it is called symbol estimation.

Turbo decoding demands for soft-output of the convolutional decoders as well. Suitable algorithms are the MAP algorithm and the SOVA (Soft Output Viterbi Algorithm).

The SOVA is usually implemented as a two-step algorithm, with a Viterbi algorithm part and a part responsible for calculating the soft-outputs. The state metric unit of the part realizing the Viterbi can be implemented based on a trace-back or a register-exchange structure. The soft-output calculation part consists mainly of a competing path

computation unit. Except for low throughputs, this unit is implemented as a register-exchange architecture. A major drawback of register exchange units is that they do not lend themselves well to hardware folding. It is therefore difficult (if not impossible) to obtain efficient SOVA architectures for a wide range of throughput requirements. Furthermore the communication performance of the SOVA with optimal soft update can be only as good as the sub-optimum MaxLogMAP algorithm. For an efficient implementation the MAP algorithm is implemented.

2.1 Turbo decoding

Decoding Turbo codes by searching the most likely codeword is far too complex. Therefore iterative decoding is advised. The two convolutional codes are decoded separately. While doing this, each decoder incorporates information that has been gathered by the other. This "gathering of information" is the exchange of soft-output values, where the bit-estimates of one unit are transformed into a priori information for the next. The decoders hence have to be soft-input soft-output (SISO) units.

The confidence in the bit estimation is represented as a Log-Likelihood-Ratio (LLR):

$$\Lambda(d_k) = \ln \frac{P(d_k = 1)}{P(d_k = 0)} \tag{2.1}$$

The sign shows whether this bit is supposed to be one or zero whereas the confidence in the decision is represented by the magnitude.

In order to extract the information that has been gathered during the last decoding stage, the systematic and a priori information that lead to this estimate have to be subtracted. This yields:

$$L^1(d_k) = \Lambda^1(d_k) - y_k^s - L^2_{\text{deint}}(d_k) \tag{2.2}$$

$$L^2(d_k) = \Lambda^2(d_k) - y_{k\,\text{int}}^s - L^1_{\text{int}}(d_k) \tag{2.3}$$

This is called the extrinsic information.

The confidence of one decoder in a bit to have a certain value biases the initial guess of the other.

Figure 2 shows such a Turbo code decoder consisting of two MAP decoders, an interleaver and a deinterleaver. Feeding the input of one decoder as a priori information input to the next enables the improvement over the decoding iterations. It also gave Turbo codes their name, as it resembles the "feedback-of-exhaust" used in combustion turbo engines. Inputs to the decoder are the received channel values (systematic, parity1 and parity2); during the very first MAP1 operation, the a priori information is set to zero.

2.2 <u>The maximum-A-Posteriori (MAP) Algorithm</u>

The name Maximum-A-Posteriori stems from the fact that the estimation of the bits is based on the whole receiver sequence. It is done after all the information is in.

Equation 2.4 shows the output of such a MAP decoder.

Bahl et al. described in [L. Bahl, J. Cocke, F. Jelinek, and J. Raviv. Optimal Decoding of Linear Codes for Minimizing Symbol Error Rate. *IEEE Transaction on Information Theory*, IT-20:284-287, march 1974] an efficient algorithm for the MAP decoder, which is based on recursions operating on the trellis in forward and backward recursion. That algorithm is commonly referred to as MAP or BCJR algorithm:

Let $R_k = (y_k^s, y_k^{p1}, L_k^2)$ denote the input of the MAP, with $\bar{R} = (R_1, \ldots, R_k, \ldots R_N)$, where N is the length of the block, then the BCJR-algorithm computes the a-posteriori probabilities (APP)

$$\Lambda(d_k) = \ln \frac{\Pr\{d_k = 1 \mid \bar{R}\}}{\Pr\{d_k = 0 \mid \bar{R}\}} \qquad (2.4)$$

for each data symbol d_k after reception of the symbol sequence $\bar{R}$.

It is computed using two probabilities: One, that the encoder has reached state S_k^m, with $m \in \{\ldots 2^M\}$ after k received symbols:

$$\alpha_k(m) = \Pr\{S_k^m \mid R_0 \ldots R_{k-1}\} \qquad (2.5)$$

and another, that the remainder of the input sequence will lead the encoder to the final state given the state $S_{k+1}^{m'}$ at time k+1:

$$\beta_{k+1}(m') = \Pr\{R_k \ldots R_N \mid S_{k+1}^{m'}\} \qquad (2.6)$$

For this, the probability of a transition from state S_k^m to $S_{k+1}^{m'}$ has to be known. It is depending on the code structure, the channel model, the extrinsic information of previous decoding steps and the received symbols R_k:

$$\gamma(S_k^m, S_{k+1}^{m'}) = \Pr\{S_k^m, S_{k+1}^{m'} \mid R_k\} \quad (2.7)$$

Using γ, α and β can be computed recursively by:

$$\alpha_k(m') = \sum_m a_{k-1}(m) \cdot \gamma(S_{k-1}^m, S_k^{m'}) \quad (2.8)$$

$$\beta_k(m) = \sum_{m'} \beta_{k+1}(m') \cdot \gamma(S_{k-1}^m, S_k^{m'}) \quad (2.9)$$

A known start and final state are necessary for the BCJR algorithm to perform optimally. If the trellis is not terminated, all states have to be assumed to have equal probability for k = N.

The a-posteriori probability itself can be expressed as

$$\Lambda(d_k) = \ln \frac{\sum_m \sum_{m'} \gamma(S_{k-1}^m, S_k^{m'}, d_k = 1) \cdot \alpha_{k-1}(m) \cdot \beta_k(m')}{\sum_m \sum_{m'} \gamma(S_{k-1}^m, S_k^{m'}, d_k = 0) \cdot \alpha_{k-1}(m) \cdot \beta_k(m')} \quad (2.10)$$

The large number of multiplications involved in the computation of the APP makes it less attractive for implementation. Therefore the MAP algorithm has to be transformed to the logarithmic domain, where it becomes the LogMAP algorithm, which increases numerical stability and eases implementation, while not degrading the error correction performance.

2.3 <u>The MAP algorithm in the logarithm domain : LogMAP</u>

The transformation of multiplications into additions is the motivation for defining the MAP algorithm in the Log-Domain. A problem is posed by the additions. Using the Jacobian logarithm, the additions are substituted by a new operator:

$$\ln(e^{\delta 1} + e^{\delta 2}) = \max{}^*(\delta 1, \delta 2) = \max(\delta 1, \delta 2) + \ln(1 + e^{-|\delta 1 - \delta 2|})$$

Similar the negative logarithm can be taken, this leads to

$$\min^*(\delta 1,\delta 2) = \min(\delta 1,\delta 2) - \ln(1+e^{-|\delta 1-\delta 2|}).$$

For more than two operands, the max* is applied recursively. Since the operator is associative, a tree-like evaluation can be employed, which is advantageous for hardware implementation. The sub-optimal MaxLogMAP algorithm is obtained by using the approximation

$$\max^*(\delta 1,\delta 2) \approx \max(\delta 1,\delta 2).$$

Using the max* operation, the recursions become:

$$\ln(\alpha_k(m')) = \max^*_m(\ln(\alpha_{k-1}(m)) + \ln(\gamma(S^m_{k-1},S^{m'}_k)), \qquad (2.11)$$

$$\ln(\beta_k(m)) = \max^*_m(\ln(\beta_{k+1}(m')) + \ln(\gamma(S^m_k,S^{m'}_{k+1})) \qquad (2.12)$$

Let $\ln(\alpha_k(m'))$ from now on be denoted as $\overline{\alpha}_k(m')$ (accordingly for β and γ), then the recursions take the form:

$$\overline{\alpha}_k(m') = \max^*_m(\overline{\alpha}_{k-1}(m) + \overline{\gamma}(S^m_{k-1},S^{m'}_k), \qquad (2.13)$$

$$\overline{\beta}_k(m) = \max^*_m(\overline{\beta}_{k+1}(m') + \overline{\gamma}(S^m_k,S^{m'}_{k+1}). \qquad (2.14)$$

Similar we get:

$$\begin{aligned}\Lambda(d_k) = {} & \max^*_{m,m'}(\overline{\gamma}(S^m_{k-1},S^{m'}_k,d_k=1) + \overline{\alpha}_{k-1}(m) + \overline{\beta}_k(m')) \\ - {} & \max^*_{m,m'}(\overline{\gamma}(S^m_{k-1},S^{m'}_k,d_k=0) + \overline{\alpha}_{k-1}(m) + \overline{\beta}_k(m'))\end{aligned} \qquad (2.15)$$

Computation $\overline{\gamma}$ of includes the estimation of channel values and the a priori information. Whereas the conventional method is quite complicated, an optimised branch metric calculation is used. Prior to transmission, every bit is subject to a transformation. Let $x_k \in \{0,1\}$ denote the (coded) bit, then the transmitted value is

$$y_k = -2 \cdot x_k + 1, \text{ hence } y_k \in \{-1,1\}.$$

Thus the actual mapping is '1' $\rightarrow$ '-1' and '0' $\rightarrow$ '1'.

There are only four different values per k in total the $\overline{\gamma}$ can take, one for every assumption $(x^s_k \in \{-1,1\} x^p_k \in \{-1,1\})$. The code-structure alone determines, which of them is assigned to which transition. After skipping constant factors and making additional algebraic transformations we get:

$$\bar{\gamma}(x_k^s = +1, x_k^p = +1) = 0$$
$$\bar{\gamma}(x_k^s = +1, x_k^p = -1) = \frac{4E_s y_k^p}{N_0}$$
$$\bar{\gamma}(x_k^s = -1, x_k^p = +1) = \frac{4E_s y_k^p}{N_0} + L(d_k) \qquad (2.16)$$
$$\bar{\gamma}(x_k^s = -1, x_k^p = -1) = \frac{4E_s y_k^p}{N_0} + \frac{4E_s y_k^p}{N_0} + L(d_k)$$

This simplifies the implementation significantly, as only two terms have to be computed from the channel and a priori data. One term can be dropped completely and the last one be computed from the first two. The scaling factor $\frac{4E_s}{N_0}$ is multiplied externally by usage of a working point.

Dynamic switching of the decoding algorithm

We refer now to figure 3 which illustrates a channel decoder according to the invention which is incorporated in the reception chain of a cellular mobile phone TP.

The encoded signal is being received by the antenna ANT and processed by the radio frequency stage ERF of the receiver. At the output of the ERF stage, the signal is converted into the digital domain by an A/D converter. The digital base band signal is then processed by a “rake” demodulator which is used generally in the case of a CDMA system.

Then, the channel decoding stage includes a channel decoder CTD according to the invention.

The processing chain comprises also a source decoding bloc DCS, which performs the source decoding treatments.

Turning now to figure 4, we see that the channel decoder CTD according to the invention comprises conventional preprocessing means MDM (Multiplexer, Interleaver, and Depunctionning unit) followed by a turbo-code decoder which can controllably implement either a LogMAP algorithm or a MaxLogMAP algorithm, depending on the value of a control signal SC.

Practically, this control signal SC controls a switch SW which activates or disactivates a ROM memory which contains a logarithmic table. More precisely, if a MaxLogMAP algorithm is used, the ROM memory is activated, whereas it is not the case if only LogMAP algorithm is used. Thus, the implementation overhead. for implementing a turbo-code decoder based on a LogMAP algorithm compared to a MaxLogMAP algorithm is small. The other overhead in silicon is below 2%.

In principle the performance of the LogMAP based TC decoder is better than the MaxLogMAP based decoder. Since the LogMAP decoder needs a very good channel state estimation in order to scale the input symbols this is only valid if this estimation is possible and available. Under bad estimation conditions the MaxLogMAP based decoder behaves much more robust and it exceeds the performance of the LogMAP based decoder. The invention proposes now to switch between the two decoder architectures based on the conditions of the channel state estimation.

One way for determining the channel state estimation conditions consists in determining the quality of the Signal to Interference Ratio (SIR). More precisely, an error information representing an error of the SIR estimation is determined and is considered as being a quality information.

Then, generally speaking, if said error information is lower than a predetermined threshold, the channel state estimation conditions are considered as being good and the LogMap algorithm is selected. On the contrary, the MaxLogMAP algorithm is selected.

And, according to the invention, this selection is made dynamically. In other words, the channel state conditions are dynamically estimated during the reception of the encoded signal and of course can be changed during this reception. Consequently, the selection between the two possible algorithms can be changed also during the reception, depending for example on the environment of the mobile phone.

According to an embodiment of the invention, said error information representing an error of the SIR estimation, can be the absolute value of the error of the SIR estimation. For example, this absolute value can be determined by calculating the variance of the SIR estimation.

Thus, the channel decoder CTD can comprise means MSR for estimating the SIR. Such means are conventional and well known by the man skilled in the art. The man skilled in the art can refer for example to the paper of the TSG-RAN Working Group 1 meeting #4 (Shin-Yokohama, Japan, April 18-20, 1999) entitled "Proposal for downlink interference measurement method".

After having estimated the SIR, the control means CTL, which can be easily implemented by software, calculate the variance of this SIR estimation and compare this variance to a predetermined threshold which is in general smaller than or equal to about 0,5 dB. Practically, such threshold may be equal to about 0,2 dB.

However, the inventors have observed that the quality of the SIR estimation, and thus the quality of the channel state estimation, can be also predicted by knowing the speed of the mobile phone. And, generally, it is easier to estimate the speed of the mobile than to calculate the variance of the SIR.

Any conventional method for estimating the speed of a mobile phone can be used.

For example, the speed of a mobile terminal can be estimated using the normalized autocovariance function of the power of the received signal.

Another known method, disclosed for example in EP 1 014 107, uses an autocorelation of the filtered power of the received signal. The man skilled in the art can also refer to EP 1 026 518.

An example of selection between LogMAP algorithm and MaxLogMAP algorithm based on a speed estimation, is illustrated on figure 5.

More precisely, after the speed estimation means MSE have estimated the speed (step 50), the speed V is compared with a first threshold Th1, for example equal to 5 km/h (step 51).

If the speed V is lower than the threshold Th1, i.e. corresponding to a pedestrian mobile, then the channel state estimation conditions are considered to be good and LogMAP algorithm is selected.

If the speed V is greater than a second threshold Th2 (equal for example to 25 km/h) (step 52) then, the speed is considered to be to high for having good estimation conditions. Thus, the MaxLogMAP algorithm is selected.

If the speed V is between the two thresholds Th1, Th2, then the delay profile of the transmission channel is considered (step 53).

The delay profile gives the number of paths of the multipath transmission channel, as well as the strength of the fingers of the rake receiver.

A variable threshold Th3 is then determined.

More precisely, if there is only two pathes, the threshold Th3 is equal to 15 km/h for example.

If there is less than two pathes, the threshold Th3 is equal to 5 km/h and if there is more than two pathes, the threshold Th3 is equal to 25 km/h (steps 530, 531 and 532).

Then, the speed V is compared with threshold Th3 (step 533).

If the speed V is lower than Th3, the LogMap algorithm is selected whereas the MaxLogMAP algorithm is selected in the contrary.

By using in particular speed estimation algorithm a dynamic switching of the decoding algorithm can be implemented. Thus, the overall turbo-decoder TC works always in the best performing mode (either LogMAP or MaxLogMAP) depending on the transmission conditions.

CLAIMS

1. Method of decoding an incident turbo-code encoded signal in a receiver, comprising a step of receiving said signal from a transmission channel, and a step of digitally turbo-code decoding, characterized by the fact that it further comprises a step of dynamically determining a quality information representative of the conditions of the channel state estimation, and a step of dynamically comparing said quality information with a predetermined criteria for defining a binary result corresponding to good or bad estimation conditions, and by the fact that the turbo-code decoding step comprises dynamically selecting (SC) a Maximum a Posteriori algorithm in the logarithmic domain, called LogMAP algorithm in presence of good conditions, or an approximation of said Maximum a Posteriori algorithm in the logarithmic domain, called MaxLogMAP algorithm, in presence of bad conditions.

2. Method according to claim 1, characterized by the fact that said determination step comprises determining an error information representing an error of the Signal to Interference Ratio estimation, called SIR estimation, said error information being said quality information, and by the fact that said predetermined criteria comprises at least one predetermined threshold (Th1, Th2, Th3).

3. Method according to claim 2, characterized by the fact that said error information is the absolute value of the error of the SIR estimation, and by the fact that said LogMAP algorithm is selected if the absolute value of said error is smaller than said threshold, and said MaxLogMAP algorithm is selected if the absolute value of said error is greater than or equal to said threshold.

4. Method according to claim 3, characterized by the fact that said threshold is smaller than or equal to about 0,5 dB.

5. Method according to claim 4, characterized by the fact that said threshold is equal to about 0,2 dB.

6. Method according to claim 2, for a receiver belonging to a wireless communication system, characterized by the fact that said receiver is mobile, by the fact that said determination step comprises a

mobile receiver speed estimation, said speed (V) being said error information.

7. Method according to claim 6, characterized by the fact that said receiver is a cellular mobile phone.

8. Method according to claim 2, for a receiver belonging to a wireless communication system and cooperating with a mobile terminal, characterized by the fact that said determination step comprises a mobile terminal speed estimation, said speed being said error information.

9. Method according to claim 8, characterized by the fact that said receiver is a base station.

10. Method according to any one of claims 6 to 9, characterized by the fact that said predetermined criteria comprises also the delay profile of the transmission channel.

11. Method according to claim 10, characterized by the fact that said predetermined criteria comprises a first predetermined threshold (Th1), a second predetermined threshold (Th2), greater than the first threshold, and a third threshold (Th3) depending on said delay profile, by the fact that if said estimated speed (V) is lower than said first threshold, the LogMAP algorithm is selected, by the fact that if said estimated speed is greater than said second threshold, the MaxLogMAP algorithm is selected, by the fact that if said speed is between the first and second thresholds, the speed is compared with said third threshold determined taking into account the delay profile, and if said estimated speed is lower than said third threshold, the LogMAP algorithm is selected, whereas if said estimated speed is greater than said third threshold, the MaxLogMAP algorithm is selected.

12. Receiver, comprising reception means for receiving an incident turbo-code encoded signal from a transmission channel, and digital processing means connected to the reception means and including demodulation means (RR) and turbo-code decoding means (CTD), characterized by the fact that said turbo-code decoding means are controllably configurable, in response to a binary control signal (SC), between a first configuration implementing a Maximum a Posteriori algorithm in the logarithmic domain, called LogMAP algorithm, and a

second configuration implementing an approximation of said Maximum a Posteriori algorithm in the logarithmic domain, called MaxLogMAP algorithm, and by the fact that said digital processing means further comprise auxiliary processing means (MSE, MSR) for determining a quality information representative of the conditions of the channel state estimation, and control means (CTL) for comparing said quality information with a predetermined criteria and delivering said binary control signal having a first value corresponding to good estimation conditions and a second value corresponding to bad estimation conditions, said turbo-code decoding means being switched in their first configuration when the binary control signal has the first value, and in their second configuration when the binary control signal has the second value.

13. Receiver according to claim 12, characterized by the fact that said auxiliary processing means comprise means for determining an error information representing an error of the Signal to Interference Ratio estimation, called SIR estimation, said error information being said quality information, by the fact that said predetermined criteria comprises at least one predetermined threshold.

14. Receiver according to claim 13, characterized by the fact that said error information is the absolute value of the error of the SIR estimation, and by the fact that said control signal has the first value if the absolute value of said error is smaller than said threshold, and the second value if the absolute value of said error is greater than or equal to said threshold.

15. Receiver according to claim 14, characterized by the fact that said threshold is smaller than or equal to about 0,5 dB.

16. Receiver according to claim 15, characterized by the fact that said threshold is equal to about 0,2 dB.

17. Receiver according to claim 13, belonging to a wireless communication system, characterized by the fact that said receiver is mobile, by the fact that said auxiliary processing means comprise speed estimation means (MSE) for determining the speed of the mobile receiver, said speed (V) being said error information.

18. Receiver according to claim 17, characterized by the fact that it is a cellular mobile phone.

19. Receiver according to claim 13, belonging to a wireless communication system and cooperating with a mobile terminal, characterized by the fact that said auxiliary processing means comprise speed estimation means for determining the speed of the mobile terminal, said speed being said error information.

20. Receiver according to claim 19, characterized by the fact that it is a base station.

21. Receiver according to any one of claims 17 to 20, characterized by the fact that said predetermined criteria comprises also the delay profile of the transmission channel.

22. Receiver according to claim 21, characterized by the fact that said predetermined criteria comprises a first predetermined threshold, a second predetermined threshold, greater than the first threshold, and a third threshold depending on said delay profile, by the fact that if said estimated speed is lower than said first threshold, the control signal has the first value, by the fact that if said estimated speed is greater than said second threshold, the control signal has the second value, by the fact that if said speed is between the first and second thresholds, the speed is compared with said third threshold determined taking into account the delay profile, and if said estimated speed is lower than said third threshold, the control signal has the first value, whereas if said estimated speed is greater than said third threshold, the control signal has the second value.

ABSTRACT

Method of decoding an incident turbo-code encoded signal in a receiver, and corresponding receiver, in particular for mobile radio systems

The method comprises a step of dynamically determining a quality information representative of the conditions of the channel state estimation, and a step of dynamically comparing said quality information with a predetermined criteria for defining a binary result corresponding to good or bad estimation conditions. The turbo-code decoding step comprises dynamically selecting (SC) a Maximum a Posteriori algorithm in the logarithmic domain, called LogMAP algorithm in presence of good conditions, or an approximation of said Maximum a Posteriori algorithm in the logarithmic domain, called MaxLogMAP algorithm, in presence of bad conditions.

Ref. :fig. 4

FIG.1

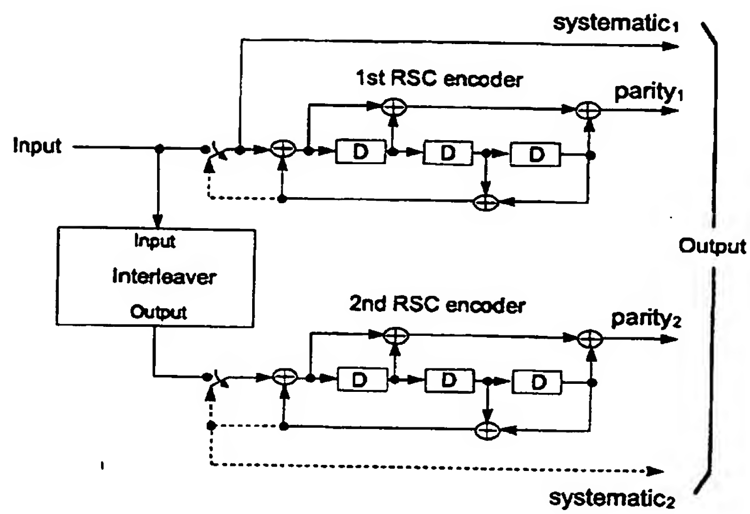

systematic1
1st RSC encoder
parity1
Input
D
D
D
Input
Interleaver
Output
Output
2nd RSC encoder
parity2
D
D
D
systematic2

FIG.2

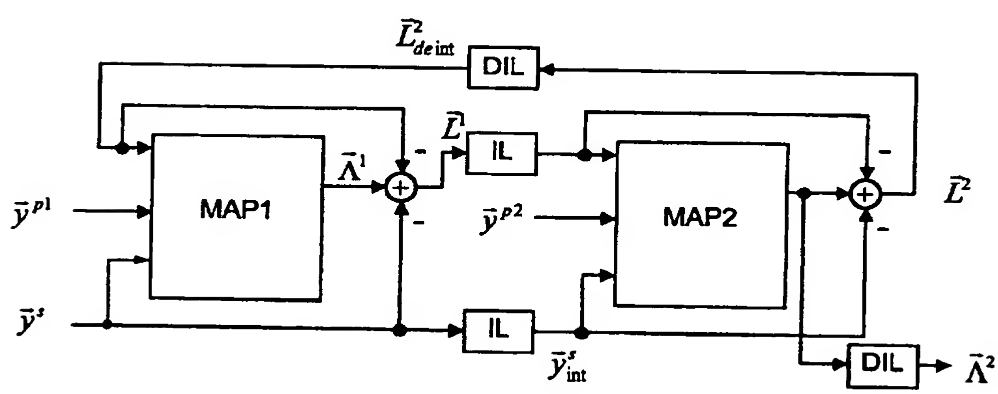

$\vec{L}^2_{deint}$
DIL
$\vec{L}$
IL
$\vec{\Lambda}^1$
MAP1
$\vec{y}^{p1}$
$\vec{y}^s$
IL
$\vec{y}^{p2}$
MAP2
$\vec{y}^s_{int}$
$\vec{L}^2$
DIL
$\vec{\Lambda}^2$

FIG.3

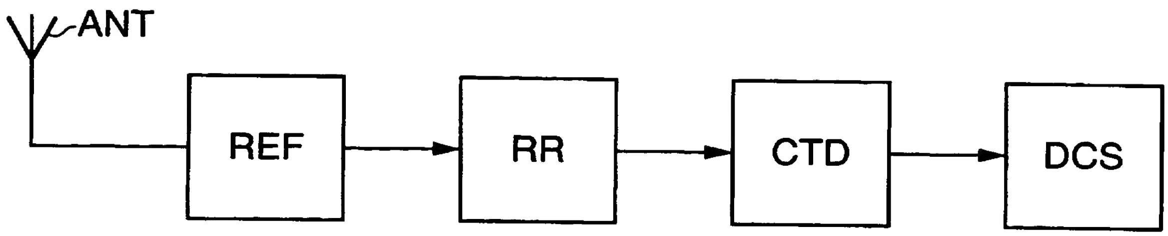

ANT
REF
RR
CTD
DCS

FIG.4

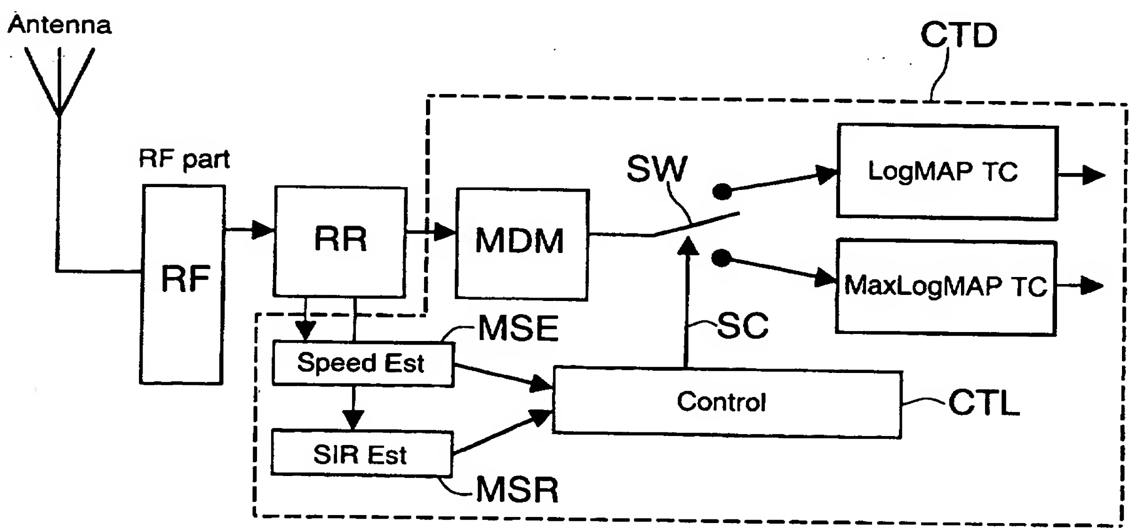
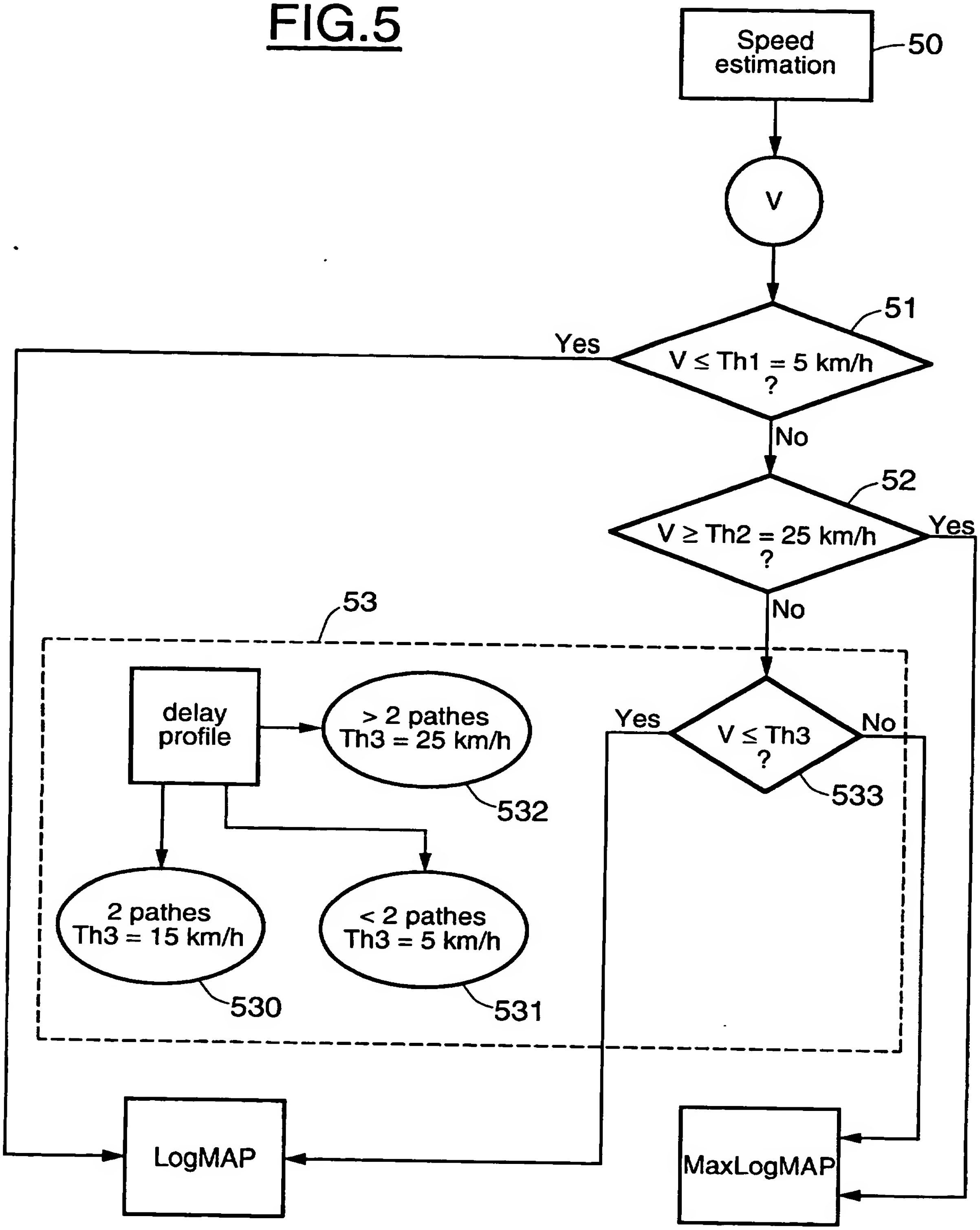

Antenna
RF part
RF
RR
MDM
SW
LogMAP TC
MaxLogMAP TC
CTD
MSE
Speed Est
SC
Control
CTL
SIR Est
MSR

FIG.5



Speed estimation
50
V
51
Yes
V ≤ Th1 = 5 km/h ?
No
52
V ≥ Th2 = 25 km/h ?
Yes
No
53
delay profile
> 2 pathes Th3 = 25 km/h
532
Yes
V ≤ Th3 ?
No
533
2 pathes Th3 = 15 km/h
530
< 2 pathes Th3 = 5 km/h
531
LogMAP
MaxLogMAP